

**DIGITAL MICROPHONE ARRAY IMPLEMENTATION
FOR HEARING AIDS PERFORMANCE IMPROVEMENT
IN A NOISY ENVIRONMENT**

An Undergraduate Research Scholars Thesis

by

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ABSTRACT

Digital Microphone Array Implementation for Hearing Aids Performance Improvement in a
Noisy Environment. (May 2014)

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Traditional hearing aids do not work well in a noisy environment. In a cocktail party scenario, we receive sound from different sources. In this case, hearing aids amplify noise and signal at the same time. As a result, the performance of hearing aids is reduced in a noisy environment. Beamforming algorithms have been applied to amplify sound from a specific direction. However, the effect of this class of algorithms is limited because the beamforming algorithms require multiple microphones.

Recent performance improvements of digital micro-electrical-mechanical system (MEMS) microphones in noise level, power consumption, and form factor make them appropriate for use in hearing aids combined with the algorithms using multiple microphones. High power noise rejection ratio of MEMS microphones eliminates the need for additional voltage regulating circuitry. In addition, digital MEMS microphones implement the mandatory analog to digital converting circuitry within the microphone. As a result, digital MEMS microphones allow the hearing aids to incorporate more microphones within a limited area.

This paper presents digital MEMS microphone array implementation for hearing aids performance improvement in a noisy environment. The microphone array system is designed using Altium Designer 10 and fabricated using 2 layer printed circuit board (PCB). This array system has four digital MEMS microphones (ADMP621 from Invensense) that are 8 cm apart. The microphones are organized as a linear array. This system also has an on board power management unit to convert input voltage to 3.3V, USB B mini power input of 5V, and internal clock of frequency 3.072MHz and duty cycle of 50%. The board needs to be powered using 6.3V to 10V voltage source or exact 3.3V voltage source. This board also has one external clock input or internal clock output SMA ports, four digital data SMA output ports for each microphone, and two digital data SMA output ports for stereo output of four microphones.

DEDICATION

God who created this world.

My parents, Jianfen Ma and Wuke Zhang.

ACKNOWLEDGMENTS

First, I would like to express my sincerely gratitude to my advisor and the supervisor of this thesis Professor Jose Silva-Martinez for his continuous support to my Undergraduate Research. He has provided me the insights to be an effective researcher. I have learned a great deal from him. I would also like to thank Alexander Edward and Carlos Briseno for their help in PCB design.

CHAPTER 1

INTRODUCTION

Hearing loss is one of the major health issues in the world. Over 5% of world's population has a disabling loss of hearing. People started to invent hearing aids to help the hearing loss community as early as the 17th century. The purpose of hearing aids is to improve the signal to noise ratio so that people can collect desired information. In other words, the hearing aids amplify the desired sound to compensate for the hearing losses of the user. However, we live in a noisy world [1]. And, traditional hearing aids could not function well in a very noisy environment [2]. In a noisy environment, the desired sound is mixed among all the unwanted noises. As a result, the hearing aids not only amplify the desired sound, but the undesired noise as well. So, how do we make hearing aids devices minimize the noise? This issue raises the challenge for the next generation hearing aids.

We need a method to amplify the desired sound source and attenuate all the other noise sources. Typically, sound source can be differentiated based on the characteristics of the sound. The characteristics of the sound wave are frequency, amplitude, wave shape and phase, though many more are related [3]. Based on these sound wave characteristics, we can separate sounds that are produced by different sources. In this way, we can eliminate the unwanted sound. However, sometimes the characteristics of the sound source and noise sources are similar. Intuitively, we can see that the benefits of the sound character based algorithms will diminish if the noise and desired signal have similar sound characteristics. Suppose two people are talking at the same time, the listener could not use the generic human speech algorithm to separate these two sounds.

It is difficult for even a person with normal hearing abilities. In order to hear one person better, the listener would naturally turn the head to the desired sound source. Sometimes, they can also put hands as cone shapes on their ears to filter out the noise sources. This natural reaction reflects that the spatial locations of different sound sources are another important factors to separate different sound sources. Beamforming algorithms utilize the spatial location information to reduce sound from unwanted direction and amplify the sound from the desired direction.

Current directional hearing aids have successfully implemented the beamforming algorithms using multiple analog microphones. However, analog microphones typically require additional circuitries such as preamplifier and analog to digital convertor. These additional circuitries consume additional power and take precious space within the hearing aids device. As a result, only a limited number of microphones can be put on hearing aids. This limitation reduces the benefits of the beamforming algorithm. With the advancement of integrated circuit, digital microphones become a viable option to substitute the traditional analog microphones and the additional circuitries. Specifically, the micro-electro-mechanical system (MEMS) based digital microphones recently overcome the power consumption and noise level requirement for use in hearing aids. This paper presents a linear microphone array system using digital MEMS microphone. The immediate purpose of this research is to evaluate the feasibility and performance of digital microphone array in hearing aids application. Ultimately, this research will enhance the life quality of people with hearing aids. In addition, this research is a continuation of a summer research project at Texas A&M University [4].

The remainder of the thesis is organized as follows. Chapter II provides background information for hearing aids signal processing using multiple microphones, which includes history, typical signal flow of hearing aids, and the beamforming algorithms. Chapter III presents a survey of current microphone technology, microphone selection considerations, different design considerations for the linear microphone array system, as well as different ways to assemble the systems for evaluation. Chapter IV provides descriptions and instructions to use the assembled microphone array systems, summary of the research, and suggestions for further works.

CHAPTER II

BACKGROUND FOR HEARING AIDS

In this chapter, I briefly review some important hearing aids history. Then, I introduce a general hearing aids signal flow. Finally, I provide background information for the multi-microphone beamforming algorithms.

History of Hearing Aids Development

The first hearing aid was created in the 17th century as a physical device called the ear trumpet [5]. It is shown in Figure 2-1.



Figure 2-1: Ear Trumpet [6]

The first electronic hearing aid was created in 1898 [5]. At that time, hearing aids suffered from sound distortion in addition to other drawbacks like bulky size and large power consumption. With the development of the carbon microphone, the battery, and the transistors, hearing aids have increased the performance significantly. By the mid-1990s, digital signal processing technology is incorporated into hearing aids as the digital integrated circuit becomes smaller and faster following Moore's Law [7]-[9].

Significant research has been done recently in the area of hearing aids. Some hearing aids have telecoil that could amplify sound from a cell phone without picking up significant amount of background noise [10]. Some hearing aids use a microphone array system to capture from certain directions [10]-[12]. Further, researchers have developed directional acoustic receiving system that emphasizes sounds of interest arriving in a direction forward of the user [13].

Hearing Aids Signal Flow

Microphone is a transducer that transfers acoustic power to electrical power. Typically, the electrical power output is a very weak analog signal that is related to the input sound pressure level (SPL). Then, the electrical output of microphone is amplified by an operation amplifier to an appropriate level for analog to digital converter (ADC) input. Then, the microphone electrical signal is converted to the digital format using ADC for further digital signal processing. Typical hearing aids digital signal processing (DSP) blocks are implemented using custom chips in order to achieve low power and real time computations. After the digital signal processing block, the processed digital signal is converted back to an analog signal using digital to analog converter (DAC). Then, the analog electrical signal could be transduced back to the sound pressure using a speaker. Note that the additional gains for the analog electrical signal are required depends on the speaker requirement. Figure 2-2 below is a typical diagram for hearing aids signal flow.

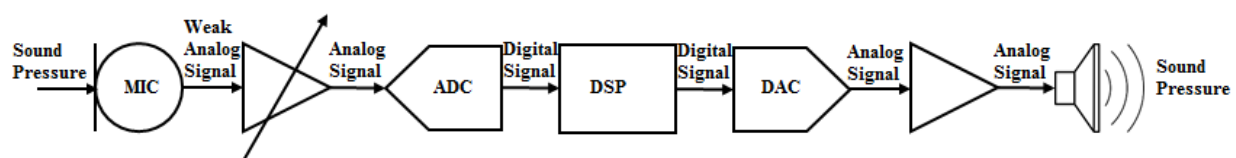


Figure 2-2: Hearing aids signal flow with analog microphone

With the advancement of integrated circuit technology, digital microphones can be manufactured smaller enough to incorporate in hearing aids while maintaining the performance. As a result, we can configure the digital microphones to directly interact with the digital signal processing block. Figure 2-3 below is a diagram for hearing aids signal flow with a digital microphone.

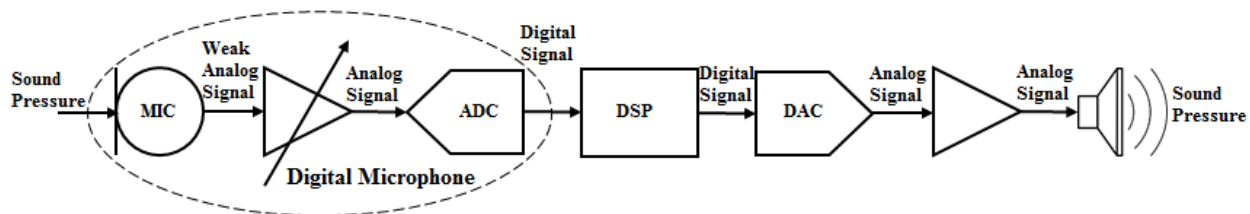


Figure 2-3: Hearing aids signal flow with a digital microphone

Beamforming Algorithms

Multi-microphone algorithms use the spatial information to increase the performance especially when the speech and the noise sources are heavily spatially separated [14]. Beamforming is a signal processing technique used in sensor arrays for directional signal transmission or reception [15]. The most fundamental beamforming technique is the delay-sum beamforming algorithm. The delay-sum beamforming algorithm uses the fact that different sound sources arrive at different microphones at different times. See Figure 2-4 for illustration. The delay-sum beamforming algorithm can perform spatial filtering to some directions assuming that the sound sources and microphones are far away from each other [16]. The delay-sum beamforming algorithm can also perform spatial filtering to a point. In general, the delay-sum beamforming algorithm adjusts internal delays to make the desired signal to arrive at the same time. In this way, the signal from the desired direction or point will be added constructively while other directions or points will have some degrees of attenuation. See Appendix A for a complete delay sum beamforming algorithm implementation using MATLAB.

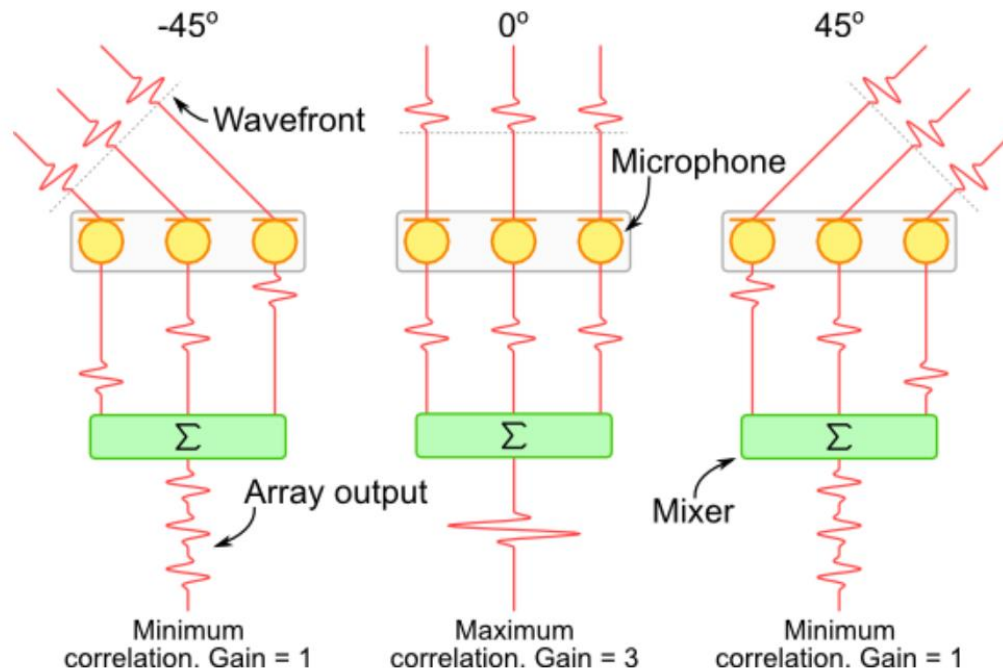


Figure: 2-4: Simple Delay-sum beamformer. If the array's output is created by summing all microphone signals, the maximum output amplitude is achieved when the signal originates from a source perpendicular to the array; the signals arrive at the same time, they are highly correlated in time and reinforce each other. Alternatively, if the signal originates from a non-perpendicular direction, they will arrive at different times so will be less correlated and will result in a lesser output amplitude [16].

If the sound characteristics of the sources are different from the noise, we can use adaptive beamforming techniques such as Griffiths-Jim Beamforming or Frost Beamforming algorithm to further enhance the performance of the spatial filter system. This idea was first introduced by Dr. Widrow [12], [17].

Almost all of the modern software tools can simulate the beamforming algorithms. The examples are MATLAB, Python, C/C++, Octave, and etc. MATLAB provides extensive libraries to support digital signal processing and simulation [18]. Beamforming algorithm can be simulated using the libraries in MATLAB [19]. The audio systems laboratory in University of Kentucky

has developed an Audio Tool Box with different beamforming algorithm implementations using MATLAB [20]. In the next chapter, we consider the design issues involved in the implementation of these techniques described in this section.

CHAPTER III

MICROPHONE ARRAY IMPLEMENTATION

While different multiple microphone based algorithms can theoretically improve the performance of hearing aids, different considerations need to be made in order to create a microphone array system satisfying different requirements such as size, power consumption, and etc. In this chapter, I present background for current microphone technology development with emphasis on digital MEMS microphone. Then, I will present the microphone selection process as well as overall printed circuit board design considerations. Finally, I will provide one possible system integration mechanism for the microphone array.

Digital MEMS Microphone

Micro-electro-mechanical systems (MEMS) technology is the technology of very small devices. Typically MEMS devices are made up of components between 1 to 100 micrometers. MEMS technology becomes a reality using the modified semiconductor device fabrication technologies, normally used to make electronics [21].

The most prevalent type of microphone is called the condenser microphone. The condenser microphone converts sound pressure information to displacement information through the vibration of the diaphragm. By attaching the diaphragm to one of two sides of a capacitor, the displacement information is transduced to electrical information for further processing.

Electret condenser microphone (ECM) technology is based on the condenser microphone and is widely used in hearing aids. Recent development of MEMS based microphone offers better

performance, manufacturing stability, lower power consumption, and higher power supply rejection ratio in addition to a small fabrication geometries. Like ECMs, MEMS microphones are designed with the condenser microphone design principles [22], [23]. Until now, the power consumption and noise levels of MEMS microphones have been too high to make them appropriate for use in hearing aids, but new devices that meet these two key specifications are now enabling the next wave of innovation in hearing aid microphones [22].

The fact that MEMS technology is fabricated in silicon makes it easy to incorporate traditionally external circuitries within the microphone package. In this way, we can reduce space and manufacturing cost for the hearing aids. Majority of the hearing aids need to process the data in the digital domain. As a result, it is natural to incorporate the audio preamplifiers and analog to digital converters within the MEMS microphones. As a result, the size of the hearing aids circuits is further reduced.

Looking back to the development history of microphone, every major performance boosts of hearing aids were enabled by the advancements of a particular fabrication technology. I believe that MEMS technology will enable the development of the next generation hearing aids.

Currently, MEMS microphone market is over \$582 million in 2012 with prediction of \$1.3 million in 2017. In 2013, 2.6 billion units of MEMS microphone is shipped with prediction of 5.4 billion prediction in 2017 [24].

Digital MEMS Microphone Selection

Microphone selections is crucial in order to obtain a good hearing aids performance. In order to have a clear understanding for microphones, I have listed some common terminologies that appear frequently in microphone datasheets. In addition, I have put some common rules for microphone selections in this section. I also present the microphone selection process from multiple digital MEMS microphones.

Below are a list of important parameters for microphone selection.

- Max Sound Pressure Level (Max SPL) is the maximum sound pressure input for a given microphone. The microphone output signal will start to distort if the given input sound pressure is above Max SPL. Since the human sound threshold of pain is 120 dB SPL, sound pressure above 120 dB SPL does not provide additional benefits for microphone.
- Input noise is the noise generated by microphones themselves. The microphones cannot pick up signal with strength less than the input noise. Smaller input noise microphones have a better performance. Input noise does not provide additional performance gain if it is below the absolute threshold of hearing because people would not tell the differences in such low sound pressure level. Typically, the threshold of hearing is above 4dB SPL.
- Signal to noise ratio (SNR) is the range between the input noise and a reference signal of 94 dB SPL. Specifically, signal to noise ratio is typically measured with a 1 kHz sine wave at 94 dB sound pressure level (SPL), or 1 pascal (Pa) pressure. Bigger SNR microphone has better performance. Note that the reference signal sound pressure level equals to SNR plus the input noise.
- Dynamic range is the range between the input noise and the Max SPL.

- Sensitivity for an analog microphone is the ratio of the analog output voltage or digital output value to the input pressure. It is typically measured with a 1 kHz sine wave at a 94 dB sound pressure level (SPL), or 1 pascal (Pa) pressure. Analog microphone sensitivity is typically specified in logarithmic units of dBV (decibels with respect to 1V). The sensitivity for digital microphone has unit dBFS (decibels with respect to digital full scale). One could not say that more sensitivity the better or the other way around. Different sensitivity levels fit different applications. Typically, sensitivity should be between -46 dBV and -35 dBV for a typical audio application. Analog Device, a leader in semiconductor industry, has an article that describes the distinction in sensitivity specifications between analog and digital microphones [25].
- Directionality describes the microphone's sensitivity to sound from various directions. Omnidirectional microphone picks up sound evenly from all directions. Unidirectional microphone picks up sound predominantly from one direction [26]. Knowles Electronics is one of the leaders for microphones design. Based on the interview that I have with a sale representative, currently the omnidirectional microphone (TM, TO, and FG series) are used commonly in hearing aids.
- Total Harmonic Distortion (THD) is the ratio of the sum of the powers of all harmonic components to the power of the fundamental frequency. THD is typically specified with around 105 dB SPL. Based on a discussion with my adviser, THD needs to be below 1% with 100 dB SPL input.

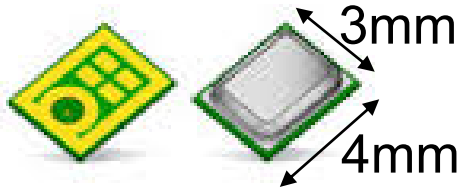


Figure 3-1: ADMP621 Microphone [27]

Three different digital MEMS microphones are evaluated. Based on the considerations on the above parameters, I selected ADMP621, a wide dynamic range microphone with PDM digital output by Invensense (Figure 3-1). Refer to table 3-1 for key parameters comparison. APMP621 has 5 pins: power, ground, clock, data select, and data output. Refer to datasheet for more details [27].

Table 3-1: Key parameters comparison for three digital MEMS microphones.

Part Name	MP34DT01 [28]	ADMP621 [27]	AKU440 [29]
Sensitivity (dBFS)	-29/-26/-23	-48/-46/-44	-28/-26/-24
SNR (dB)	63	65	63
Dynamic Range (dB)	89	111	89
THD worse case<1% (pay attention to input SPL)	<1% if 100dBSPL <5% if 115 dBSPL	0.35% / 1% if 105 dBSPL	<1% if 100dBSPL <5% if 110 dBSPL
Type of ADC	Cannot find	4 th order sigma delta modulator	NA
Clock (MHz)	1 / 2.4 /3.25	1/3.072/3.6	1/--/3.25
Supply Voltage (V)	1.64 / 1.8 / 3.6	1.62 / -- / 3.63	5.5
Noise Margin	0.35 / 0.65	0.3 / 0.7	NA

Printed Circuit Board Implementations

I used Altium Designer 10 for the printed circuit board (PCB) design of the digital MEMS microphone array. Getting started tutorial for Altium Designer is available online [30]. I fabricated the PCB through Advanced Circuit \$33 dollar student special [31]. Note that the largest dimension allowed with this price is 10 inches. I purchased the majority of the printed circuit board components through Mouser [32]. I used ARQ Electronics, a local Electronics Company at College Station, Texas, to assemble the major PCB components such as the digital MEMS microphones and surface mount chips [33]. For the remainder of this chapter, I describe the design of two digital MEMS microphone array systems.

I implemented two printed circuit boards. The purpose of the first board is to get data output from a single digital MEMS microphone using oscilloscope. The first PCB consists of one digital MEMS microphone, a mini USB power input, a voltage regulation unit, and an internal clock unit. The board has the size of 5cm by 5cm dimensions and input voltage of 1.8V. See Figure 3-2 for detailed component placement. This board can take an external clock input and output received microphone clock and the microphone digital data using SMA ports. This board has two layers. I routed both the power and signal lines at the top layer. I poured ground plane to both top and bottom layers. I specified the power signal wire width to be from 1mm to 2mm and other signal wire width to be from 0.3mm to 1mm. Jumpers are needed for this PCB to operate. In the P1 header, pin2 and pin4 need to be connected in order to power up the board. In the P2 header, pin2 and pin4 need to be connected in order to have the digital microphone to receive the internal clock. Pin1 and pin2 need to be connected in order to have the digital microphone receive the external clock. A LED is placed on board to indicate if the board is powered properly.

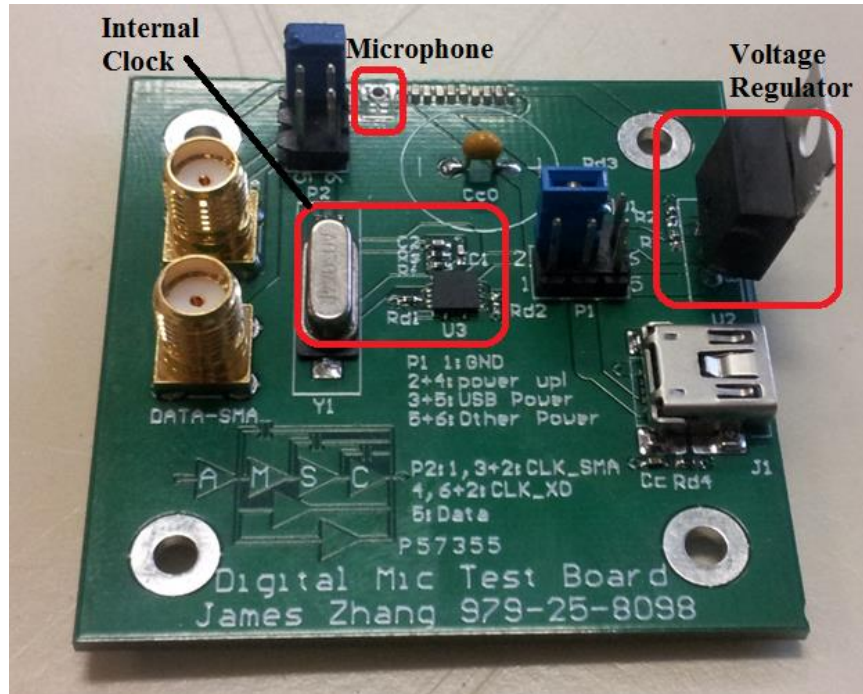


Figure 3-2: First stage PCB design with single Digital MEMS microphone

The second PCB consists of four digital MEMS microphones arranged as a linear array with 8cm apart, a mini USB power input, a voltage regulation unit, and an internal clock unit. The board has the size of 254mm by 82.5mm and the input voltage of 3.3V. This board can take an external clock input and output each microphone clock and digital data through SMA ports. In addition, this board allows to transmit two data lines using one output SMA port using stereo mode specified in the datasheet [27]. This board has two layers. I routed the signal line at the top layer and power line at the bottom layer. I poured ground plane to both top and bottom layers. I specified the power signal wire width to be from 2mm to 3mm and other signal wire width to be from 0.2mm to 1mm. In order to ease the debugging effort, I added LED indicators for power, internal clock, external clock, data selects for all four microphones, and data outputs for all four microphones. In order to make this board be more user friendly with both the digital signal processing unit using serial data transmission protocol and the oscilloscope, I decided to use four

microphones for the linear microphone array system. Typically oscilloscope has only 4 ports, I hope to monitor all ports at the same time. In addition, I am planning to use beagle bone black [34] serial peripheral interface bus (SPI) to transmit data for further process and there are only two available SPI buses. With all microphones configured to output stereo mode as specified in the datasheet, a maximum of four data outputs can be recorded at the same time. Based on the PCB manufacture, the maximum dimension allowed for the student price PCB is 10 inches. As a result, I made the distance between each microphone 8cm apart in order to achieve the maximum yield for the spatial filtering effect. Table 3-2 summarizes all the header connections for this board. Figure 3-3 is the top layer layout view for PCB. Refer to Appendix B for complete design documents for the linear microphone array design.

Table 3-2: Summary of jumper configuration for linear microphone array board

Header name	Pin Connection	Description
P_power	2 and 4	Connect the power supply unit with the remainder of the PCB circuits
P_DS1,P_DS2,P_DS3,P_DS4	1 and 2	Connect Data Select to Power. Output microphone data as left channel.
	2 and 3	Connect Data Select to Ground. Output microphone data as right channel.
P_DATA12	1 and 2	Connect Microphone 1 to DATA12 output
	5 and 6	Connect Microphone 2 to DATA12 output
P_DATA34	1 and 2	Connect Microphone 3 to DATA34 output
	5 and 6	Connect Microphone 4 to DATA34 output
P_clock	1 and 2	Select External Clock input
	2 and 4	Select Internal Clock input

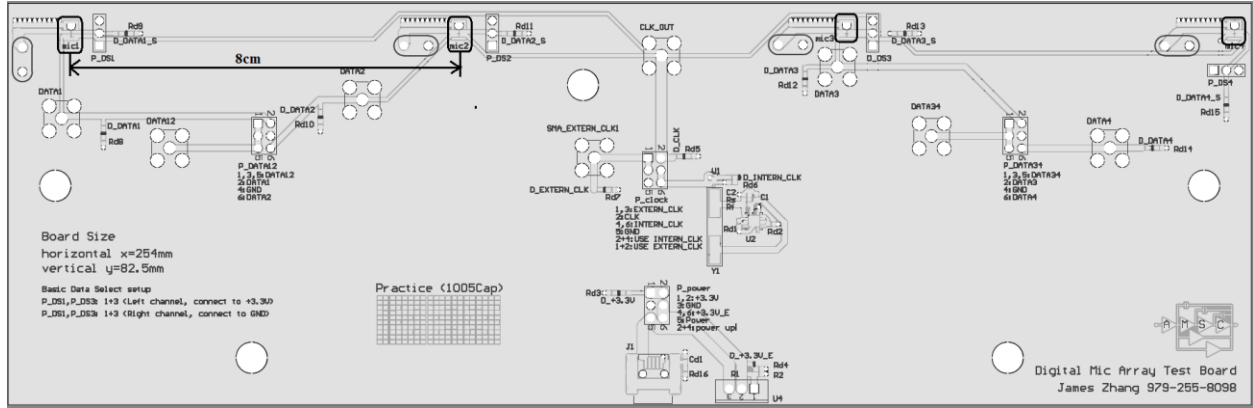


Figure 3-3: linear microphone array top layer layout

Both PCB boards utilize a linear voltage regulator, an internal clock, and multiple LED indicators. I will use the space below to describe the fundamental design principles for each unit. Then, I will present considerations for coupling capacitance selection, component placement, and custom footprint design.

A linear voltage regulator is added to protect the circuit from high voltage input and maintain a steady input voltage. The Digital MEMS microphone can only take a supply voltage between 1.62V to 3.63V. If the supply voltage is below 1.62V, the microphone will not work properly. If the supply voltage is above 3.63V, the microphone will be damaged. I selected the 3-terminal adjustable regulator LM317 (KCT) from Texas Instrument for the power management unit design [35]. The key datasheet information is that the minimum input and output voltage difference is 3V. The key design schematic is shown in the Figure 3-4. The output voltage can be calculated using $V_o \approx V_{ref} \left(1 + \frac{R_2}{R_1}\right)$ where $V_i - V_o > 3V$. For the 1.8V input power single microphone board design, I choose $R_1 = 750\Omega$ and $R_2 = 330\Omega$. For the 3.3V linear microphone array design, I choose $R_1 = 732\Omega$ and $R_2 = 1200\Omega$.

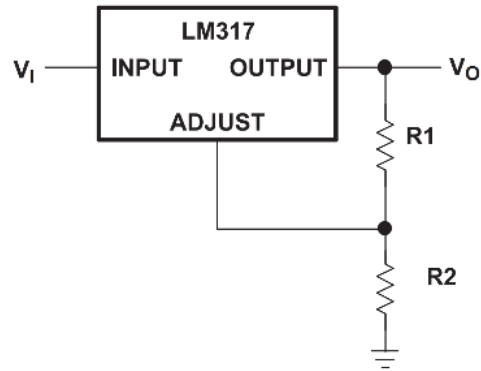


Figure 3-4: 1.25-V to 20-V Regulator Circuit

We need to have an input clock signal to the digital MEMS microphone. According to the datasheet, we need the input clock signal to have frequency from 1 MHz to 3.6 MHz with duty cycle from 40% to 60%. My goal is to design an internal clock with 3.072 MHz frequency and 50% duty cycle since the digital microphone is tested under this input clock condition. In order to obtain robust performance with respect to various environmental variables, I decide to use Pierce Gate Crystal Oscillator design [36]. Since the crystal oscillator duty cycle output varies more than the specified input condition, I decide to use crystal oscillator with 6.144 MHz frequency, which is twice of the required internal clock output frequency. Then, I used a flip flop to reduce the clock cycle to 3.073 MHz. In this way, I can guarantee that the duty cycle is almost 50% [37]. I used a flip flop (SN74AUP1G80) unit from Texas Instruments [38]. I also used an inverter (SN74AUC14) from Texas Instruments [39]. I used the 6.144 MHz crystal oscillator with 18pF load capacitance from ECS Inc. [40].

Based on the design documents, I have found values for additional resistors and capacitors.

Figure 3-5 contains the clock design with component values. Note that R_{d1} and R_{d2} are simply pull down resistors to tie the unused pins to ground.

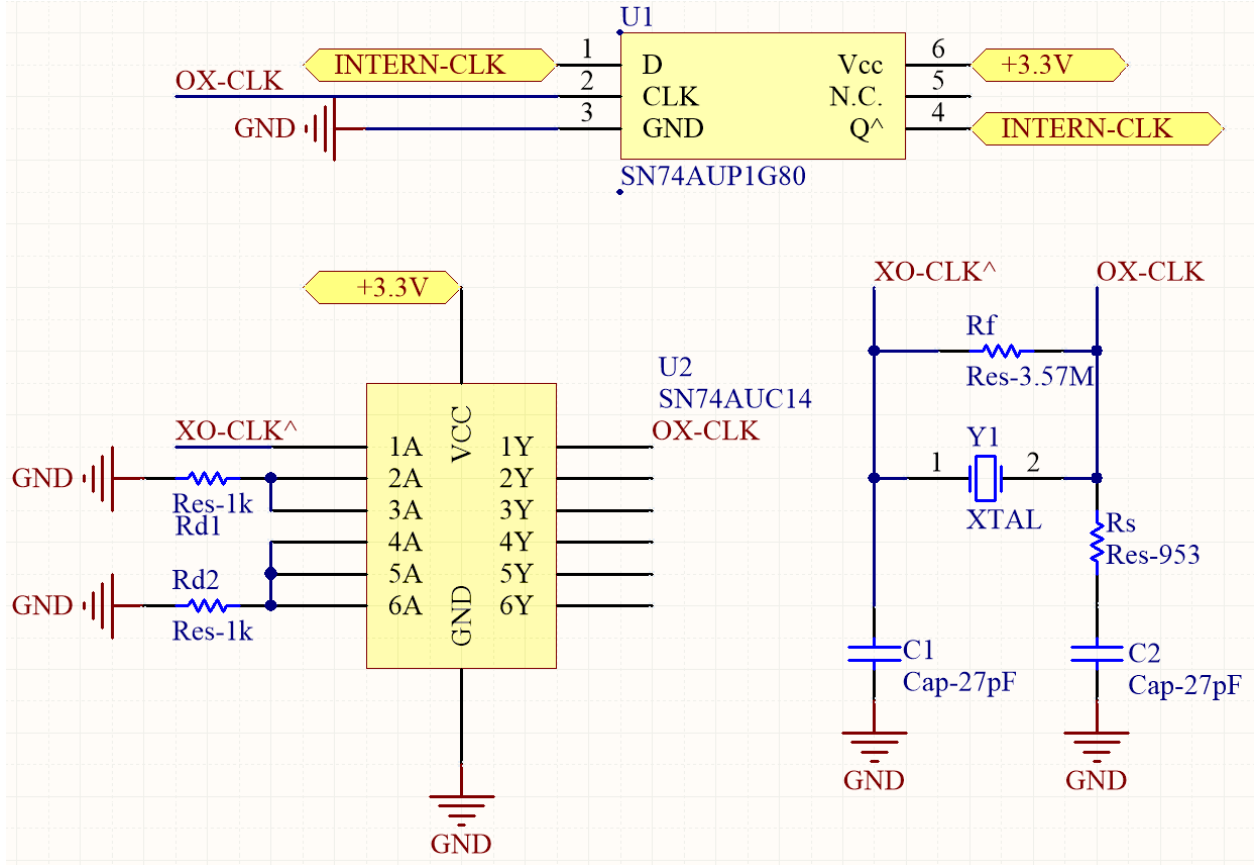


Figure 3-5: Schematic Design of Internal Clock

LED also needs proper current input to function. Typically, a bias resistor is added in order to adjust the current through the LED. (See Figure 3-6 below) This bias resistor value can be

calculated using equation $R = \frac{V_{supply} - V_{forward}}{I_{forward}}$. I used LED from ROHM semiconductor [41].

According to the datasheet, the typical forward current is 5mA and the typical forward voltage is

2.9V. Since the input power is 3.3V, we need to have a bias resistance of value $\frac{3.3V - 2.9V}{5mA} = 80\Omega$.

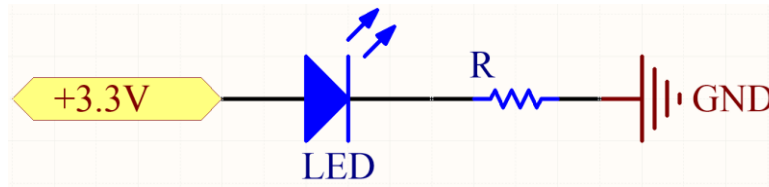


Figure 3-6: Typical LED design

In addition to the major units design, coupling capacitors selection and placement are considered in both board designs to maximize performance and usability. Board noise refers to the imperfection of input power of the microphone. Board noise can be caused by inherence noisy power input, environment effect, board layout, and power usage. Coupling capacitors smooth out these imperfections. However, a certain capacitor is only effective within a limited range of frequency. Further, coupling capacitors need to be placed close to the device in order to achieve maximum noise reduction effect. Two types of capacitors are used in the PCB board designs to reduce input power noise. Around each microphone, I put one capacitor of value $0.1\mu F$ that reduces noise within a frequency of 10 MHz [42] and ten capacitors of value $10\mu F$ that reduces noise within a frequency of 1.1 MHz [43]. In order to achieve better usability, SMA connectors and pin headers are placed around each corresponding microphone. I also make sure that each components have enough space between for easier use.

While designing the microphone foot print, we need to put special caution to make sure that each pin is mapped correctly. Typically, datasheet provides the pin label as if we are looking at the chip pins. However, footprint pin design is the mirror image of the datasheet specifications.

Refer to Figure 3-7 for illustration.

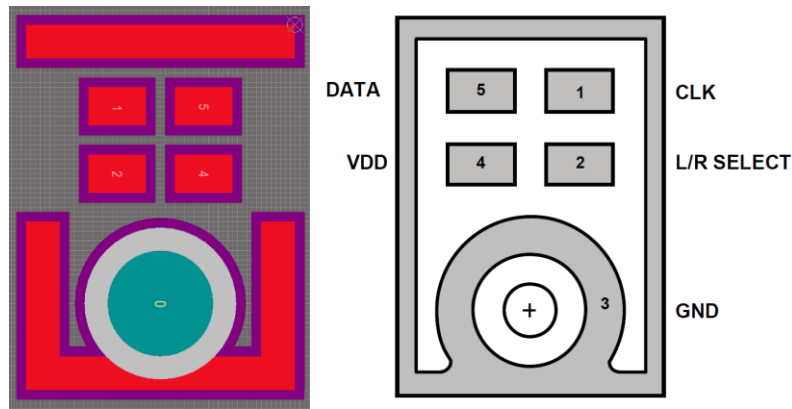


Figure 3-7: ADMP621 digital MEMS microphone footprint (left) and pin specifications (right)

Connecting the Microphone Array to Digital Signal Processing Unit

While we can use the oscilloscope to see the data outputs from the digital MEMS microphone array system, we need find ways to connect the output of microphone data to digital signal processing unit for further process. The first step would be to find a way to store the data in a memory block. I used Serial Peripheral Interface Bus (SPI) ports in Beagle Bone Black, an embedded computer, to connect with the microphone data outputs.

The digital MEMS microphone output data is in the form of the Pulse Density Modulation (PDM) [44]. PDM is just oversampled 1-bit Pulse Code Modulation. Since the Data output of the microphone is a loci-level signal typically switching at around 3MHz with fast edges. It therefore needs to be treated with the same care as any other fast signal (such as SPDIF, or analog video). It is important to use good quality coax cable and to terminate the signal correctly.

CHAPTER IV

SUMMARY AND DISCUSSION

A digital MEMS microphone test board and a digital MEMS microphone array system are developed. Table 4-1 summarizes the key specifications for the digital MEMS microphone test board. Table 4-2 summarizes the key specifications for the digital MEMS microphone array system.

Table 4-1: Digital MEMS microphone test board summary

Dimensions	40mm,40mm
Supply Voltage using power management unit	4.8V to 10V
Direct Supply Voltage	3.3V
Internal Clock	3.072MHz
Microphone Counts	4
Microphone Distance	8 cm

Table 4-2: Digital MEMS microphone array system summary

Dimensions	254mm,82.5mm
Supply Voltage using power management unit	6.3V to 10V
Direct Supply Voltage	3.3V
Internal Clock	3.072MHz with 50% duty cycle
Microphone Counts	4
Microphone Distance	8 cm

In Chapter I, I describe the needs for high performance small microphones in order to fully realize the benefits of the beamforming algorithms in hearing aids in a noisy environment. I propose to evaluate digital MEMS microphones for possible usage in hearing aids application. I articulate that digital MEMS microphones have high performance, low power cost, small form factors, and can eliminate mandatory external circuitries due to the incorporation of the preamplifier and the analog to digital converter (ADC) within the microphone chip.

In Chapter II, I provide a brief history of hearing aids technology. I highlight that each hearing aid performance increase is associated with a major technology revolution. I predict that the MEMS technology will increase the performance of hearing aids in the near future. Then, I provide a general signal flow diagram for hearing aids device. Finally, I provide the basic knowledge for the beamforming algorithm.

In Chapter III, I provide background for MEMS technology and digital MEMS microphones. Then, I introduce specific microphone selection criteria, and the digital MEMS microphone array design. Finally, I discussed possible digital MEMS microphone array system interface with digital signal processing block.

Currently, I am assembling the components for the digital MEMS microphone array system. While assembling the components, I need to make sure that all the electrical connections correspond with the design (See figure 3-3 for pin headers and component locations). Then, I need to make sure that the internal units such as the USB power, the power management unit, and the internal clock are working properly. In order to test USB power, I need to use USB cord to connect the board with USB power source and test the voltage difference between pin 3 (ground) and pin 5 (power) of the P_power header. The voltage difference should be between 4.75V and 5.25V. In order to the test power management unit, I must supply voltage over 6.3V to pin 5 of the P_power header and test the voltage difference between pin 3 (ground) and pin 4 (+3.3V) of the P_power header. The voltage difference should be around 3.3V. In order to test

internal clock, I need to properly power the board using 3.3V. Then, I need to connect pin 3 and pin 4 of the P_clock header. Then, I need to direct the Internal Clock signal using SMA_EXTERN_CLK1 port to an oscilloscope. The internal clock output should have frequency of 3.072 MHz and duty cycle of 50%.

After examining all the supporting units above, I need to ensure that each microphone works. Select pin configurations described in Table 4-3 below. Then, connect DATA1, DATA2, DATA3, DATA4, and SMA_EXTERN_CLK1 ports to an oscilloscope to see if data can be captured as a series of logical 0s and 1s. Based on the configurations described in Table 4-3, DATA1 and DATA3 should be captured in the falling edge of the clock signal. DATA2 and DATA4 should be captured in the rising edge of the clock signal.

Then, I need to make sure that the stereo mode for the microphone array is working properly. Table 4-3 below only used one possible stereo combination option. Note that never select both D_DATA1_S and D_DATA2_S to ground or power. If the pin configuration for D_DATA1_S is pin1 connect to pin2, D_DATA2_S pin configuration must be pin2 connect to pin 3. This rule applies for D_DATA_3_S pin header and D_DATA4_S pin header selection as well. In order to see if stereo mode is working properly for all microphones, connect DATA12, DATA34, SMA_EXTERN_CLK1 ports to an oscilloscope to see if DATA12 and DATA34 can be captured in both rise and falling edges of the supplied clock.

Table 4-3: pin configuration for microphone array testing. The configuration below uses external power source, external clock signal with microphone 1 and 3 sample on the falling edge of the clock and microphone 2 and 4 sample on the rising edge of the clock.

Header Name	Pin Connection	Description
P_power	2 and 4	Connect power
P_clock	1 and 2	Select External Clock input
P_DS1, P_DS3	1 and 2	Connect Data Select to Power. Output microphone data as left channel.
P_DS2, P_DS4	2 and 3	Connect Data Select to Ground. Output microphone data as right channel.
P_DATA12	1 and 2	Connect Microphone 1 to DATA12 output
	5 and 6	Connect Microphone 2 to DATA12 output
P_DATA34	1 and 2	Connect Microphone 3 to DATA12 output
	5 and 6	Connect Microphone 4 to DATA12 output

In order to utilize the system designed in this thesis, further research is needed to receive data using digital signal processing blocks (See Figure 2-4). Specifically, I need to find ways for Beagle Bone Black to take the microphone data input using SPI and store the data to memory. Since the majority of the signal processing algorithms are only compatible to the pulse coded modulation (PCM) signal, I also need to have a program to convert the PDM modulation data to the PCM modulation data. Ultimately, this board can be incorporated into an audio test to collect data for the hearing aids beamforming algorithm research.

The linear digital microphone array system has advantages such as robust performance and clear interface. However, these advantages have trade off relationship with flexibility. The distance and configuration of the microphone array are fixed. As a result, the system could not be used to

test any other spatial configurations. In order to maximize the flexibility of the system, I recommend to construct small form factor PCB board that only have one microphone with two SMA connectors for data and clock and three pin connectors for ground, power, and L/R select. In this way, we can construct different spatial configurations to fit the needs of the experiment.

Furthermore, different types of microphones can be evaluated efficiently this way. In order to construct the board in a cheap rate, I recommend to design multiple copies of PCB within one 6 inches by 10 inches space. Then, drill holes between each copy. In this way, a big PCB board can be broken to multiple smaller pieces. This method will reduce the cost of PCB manufacturing significantly.

This thesis limits the scope to only omnidirectional microphones. With the flexible configurations described above, there might be interesting research results to enhance the beamforming algorithms if we investigate the effect of directional microphones with different beam patterns.

It is also worth to note that the size limitation stressed in this thesis does not apply to non-traditional hearing aid devices that use additional microphones. In order for the beamforming algorithms to work efficiently, we not only need multiple microphones, but also enough distances between each microphones. Some non-traditional hearing aid devices have been developed to have relative large size in order to create distances between microphones. In this case, the digital MEMS microphones' small form factor advantage is not significant because the

design room is relaxed. However, small form factor digital MEMS microphones still maintain other advantages such as power consumption, performance, power rejection ration, and etc. compared to the traditional analog microphone configurations.

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APPENDIX A: DELAY-SUM BEAMFORMER

Below are the scripts that simulates delay-sum beamformer effect if the location of the sound source and noise are given. I assume the sound are recorded using two microphones and sound noise, noise source, and two microphones are arranged as the corner of a square with 60cm apart. In order to run this program, make sure that configuration.m, delay_sum_beamformer.m, and test_delay_sum_beamformer.m, mic1.wav, mic2.wav are in the same directory. Then, run the configuration.m file.

configuration.m

```
%% Initialization
clc;
clear all;
close all;

v=340; %speed of sound is approximately 340m/s

mic1='mic1.wav';
mic2='mic2.wav';

sound_enable=0;
plot_enable=1;
run('test_delay_sum_beamformer.m');
```

delay_sum_beamformer.m

```
function [ y ] = delay_sum_beamformer( x, x_prev, fs, s, m, c)
%delay_sum_beamformer beamform to a point in 3D
% Reference from work of Phil Townsend
% http://www.engr.uky.edu/~donohue/audio/Arrays/dsb.m
%
% *Inputs*
% * |x| - Matrix of raw audio data, each column a mic array track
% * |x_prev| - Matrix of audio data from a previous window. This
% data allows for real data to be padded at the front of each
% shifted track in the DSB algorithm. Simply supply zeros if
% no previous data exists.
% * |fs| - Audio sampling rate (Hertz)
% * |s| - 3x1 source location point (col vector) (meters)
% * |m| - column matrix of 3D mic positions (meters)
% * |c| - speed of sound (meters/sec)
%
% *Outputs*
%
% * |y| - Beamformed output track

narginchk(6,6);
```

```

if isempty(x) || ~isreal(x) || ~all(all(isfinite(x)))
    error('x must be a real matrix')
elseif isempty(x_prev) || ~isreal(x_prev) || ~all(all(isfinite(x_prev)))
    error('x_prev must be a real matrix')
elseif isempty(fs) || ~isreal(fs) || ~isscalar(fs) || ~isfinite(fs) ...
    || fs <= 0
    error('fs must be positive real scalar');
elseif isempty(s) || ~isreal(s) || ~all(isfinite(s)) || size(s,1) ~= 3 ...
    || size(s,2) ~= 1 || length(size(s)) ~= 2
    error('s must be a real 3x1 vector');
elseif ~isreal(m) || ~all(all(isfinite(m))) || size(m,1) ~= 3
    error('m must be a real matrix with 3 rows')
elseif isempty(c) || ~isreal(c) || ~isscalar(c) || ~isfinite(c) || ...
    c <= 0
    error('c must be positive real scalar');
end

% sample_size is number of samples in each audio track
sample_size = size(x,1);
% microphone_size is number of microphones in the array
microphone_size = size(x,2);

if microphone_size ~= size(m,2),
    error('microphone size does not agree with each other (x,m)');
end
% initialize shifted track matrix
x_delay = zeros(sample_size,microphone_size);

dist = sqrt(sum((m-s*ones(1,microphone_size)).^2,1));

t = (max(dist)-dist)/c;

% Convert time delay to digital delays
delays = round(t*fs);

% apply delay
for k=1:microphone_size
    x_delay(:,k) = [x_prev(end-delays(k)+1:end, k);
                   x(1:end-delays(k), k)];
end

% Sum the delayed tracks and save as the beamformer output
y = sum(x_delay, 2);
end

```

test_delay_sum_beamformer.m

```

%=====
% Author: Zhenxuan "James" Zhang
% Contact: zhenxuanjameszhang@tamu.edu
% Adviser: Dr. Jose Silva-Martinez
% Description: this file contains the source code for beam forming
%=====

```

```

%% Read all the microphone wav file.
[y1, fs1] = audioread(mic1);
[y2, fs2] = audioread(mic2);

player1 = audioplayer(y1,fs1);
player2 = audioplayer(y2,fs2);

% plot the frequency spectrum of microphone 1
% reference http://www.mathworks.com/matlabcentral/answers/36430-plotting-the-frequency-spectrum
N = size(y1,1);
Y1=fftshift(fft(y1));
df = fs1/N;
f= -fs1/2:df:fs1/2-df;

if plot_enable==1
    figure;
    mic1_freq=plot(f,abs(Y1)/N);
    xlabel('Frequency (in hertz)');
    title('Magnitude Response of mic1');
end

% Input Microphone position
mic_position1=[0;0;0];
mic_position2=[0.6;0;0];
mic_position=[mic_position1,mic_position2];

source_position1=[0;0.6;0];

[result1]=
delay_sum_beamformer([y1,y2],[zeros(size(y1)),zeros(size(y2))],fs1,source_pos
ition1,mic_position,v);

player3 = audioplayer(result1,fs1);

source_position2=[0.6;0.6;0];
[result2] =
delay_sum_beamformer([y1,y2],[zeros(size(y1)),zeros(size(y2))],fs1,source_pos
ition2,mic_position,v);

player4 = audioplayer(result2,fs1);
if sound_enable==1
    playblocking(player1);
    playblocking(player2);
    playblocking(player3);
    playblocking(player4);
end

```


APPENDIX B DIGITAL MICROPHONE ARRAY DESIGN DOCUMENTS

Design documents are available upon request. Contact me at zhenxuanjameszhang@gmail.com.